WHAT IS CLAIMED IS:

1. A method for adjusting a sender rate in a packet communication system to support congestion control between a server and a client, the method comprising the steps of:

(a) transmitting a plurality of data packets to said client;

(b) determining by said client whether one of said data packets is lost over a communication connection from said server to said client;

(c) transmitting a response packet for retransmission by said client if one of said data packets is lost;

(d) computing a new sender rate based on a round-trip time (RTT) corresponding to a latency between sending said response packet to said server and receiving the corresponding retransmission of said lost packet from said server; and,

(e) transmitting said new sender rate to said server if a predetermined number of said RTTs is detected thereafter during said communication connection.

2. The method of claim 1, wherein said RTT is determined according to the following steps:

transmitting a first packet having an RTT sequence number to said server if one of said data packets is lost;

receiving a second packet containing said lost packet in response to said first packet from said server; and,

calculating said round-trip time(RTT) based on a time delay between said first packet

and said second packet.

3. The method of claim 1, wherein said communication connection between said server and said client comprises at least one of a wireless communications link, a wired communication link, and the combination of a wired communication link and a wireless communications link.

4. The method of claim 1, further comprising the steps of:

including by said client a number of acknowledgment messages, in response to the plurality of said data packets, said new sender rate specifying a transmission rate at which said server may transmit subsequent data packets to said client; and,

adjusting by said server, in response to said acknowledgment messages, said new sender rate at which said server sends subsequent data packets to said client.

5. The method of claim 1, further comprising the steps of:

including a field in said response packet an RTT sequence number and said new sender rate; and,

determining by said client that one of said data packets is lost if said RTT sequence number received from said server is out of order.

6. The method of claim 1, further comprising the steps of:

including a field in said response packet a control action (CA) sequence number indicating the transmission of said new sender rate to said server; and,

adjusting, by said server, said new sender rate if said predetermined number of said RTTs is detected thereafter.

7. The method of claim 1, wherein said response packet is one of a negative acknowledgment (NACK) packet and a control action (CC) packet indicating the transmission of said new sender rate to said server.

8. The method of claim 1, wherein said computation of said new sender rate is based on a packet loss ratio.

9. A method for exchanging a plurality of messages between a server and a client over a communication link to support congestion control therebetween, the method comprising the steps of:

(a) transmitting a plurality of data packets from said server to said client;

(b) transmitting, by said client, a negative acknowledgment (NACK) packet for retransmission if one of said burst packets is lost;

(c) calculating, by said client, a round-trip time (RTT_i) corresponding to a latency between sending said NACK packet to said server and receiving the corresponding retransmission of said lost packet from said server;

(d) determining a new sender rate based on said calculated RTT indicating a transmission rate at which said server may transmit subsequent data packets to said client;

(e) successively transmitting a number of response packets responsive to the plurality of said data packets containing said new sender rate; and,

(f) adjusting, by said server, said new sender rate if said RTT is calculated more than a predetermined threshold value.

10. The method of claim 9, wherein said RTT is determined according to the following steps:

transmitting a first packet having an RTT sequence number to said server if one of said data packets is lost;

receiving a second packet containing said lost packet in response to said first packet from said server; and,

calculating said RTT based on a time delay between said first packet and said second packet.

11. The method of claim 9, wherein said communication link between said server and said client comprises at least one of a wireless communications link, a wired communication link, and the combination of a wired communication link and a wireless communications link.

12. The method of claim 9, further comprising the steps of:

including, by said client, a number of acknowledgment messages, in response to the plurality of said data packets, said new sender rate specifying a transmission rate at which said server may transmit subsequent data packets to said client; and,

adjusting by said server, in response to said acknowledgment messages, said new sender rate at which said server sends subsequent data packets to said client.

13. The method of claim 9, further comprising the steps of:

including a field in said response packet a RTT sequence number and said new sender rate; and,

determining by said client that one of said data packets is lost if said RTT sequence number received from said server is out of order.

14. The method of claim 9, further comprising the steps of:

including a field in said response packet a control action (CA) sequence number indicating the transmission of said new sender rate to said server; and,

adjusting, by said server, said new sender rate if said predetermined number of said RTTs is detected thereafter.

15. The method of claim 9, wherein said response packet is one of a negative acknowledgment (NACK) packet and a control action (CC) packet indicating the transmission of said new sender rate to said server.

16. A system for adjusting a sender rate in a packet communication system to support congestion control between a server and a client, comprising:

means for receiving a plurality of data packets;

means for determining whether one of said data packets is lost during transmission;

means for requesting that any lost frame packets be retransmitted;

means for computing a new sender rate based on a round-trip time (RTT) corresponding to a latency between requesting retransmission of said lost frame to said server and receiving corresponding retransmission of said lost frame from said server; and,

means for notifying said new sender rate to said server if said RTT is calculated more than a predetermined threshold value.

17. The system of claim 16, wherein said RTT is determined according to the following steps:

transmitting a first packet having an RTT sequence number to said server if one of said data packets is lost;

receiving a second packet containing said lost packet in response to said first packet from said server; and,

calculating said round-trip time(RTT) based on a time delay between said first packet and said second packet.

18. The system of claim 16, wherein said first packet includes said new sender rate specifying a transmission rate at which said server may transmit subsequent data packets to said client and an RTT sequence number, and wherein one of said data packets is determined to be lost if said RTT sequence number received from said server is out of order.

19. The method of claim 16, wherein said first packet includes a control action (CA) sequence number indicating the transmission of said new sender rate to said server.

20. The method of claim 16, further comprising means for adjusting said new sender rate at which said server sends subsequent data packets to said client.

21. A system for exchanging a plurality of messages between a server and a client over a communication link to support congestion control therebetween, comprising:

means for transmitting a plurality of data packets from said server to said client;

means for transmitting, by said client, a negative acknowledgment (NACK) packet for retransmission if one of said burst packets is lost;

means for calculating, by said client, a round-trip time (RTT_i) corresponding to a latency between sending said NACK packet to said server and receiving the corresponding retransmission of said lost packet from said server;

means for determining a new sender rate based on said calculated RTT indicating a transmission rate at which said server may transmit subsequent data packets to said client;

means for successively transmitting a number of response packets responsive to the

plurality of said data packets containing said new sender rate; and,

means for adjusting, by said server, said new sender rate if said RTT is calculated more than a predetermined threshold value.

22. The system of claim 21, wherein said RTT is determined according to the following steps:

transmitting a first packet having an RTT sequence number to said server if one of said data packets is lost;

receiving a second packet containing said lost packet in response to said first packet from said server; and,

calculating said round-trip time(RTT) based on a time delay between said first packet and said second packet.

23. The system of claim 21, wherein said first packet includes said new sender rate specifying a transmission rate at which said server may transmit subsequent data packets to said client and an RTT sequence number, and wherein one of said data packets is determined to be lost if said RTT sequence number received from said server is out of order.

24. The system of claim 21, wherein said first packet includes a control action (CA) sequence number indicating the transmission of said new sender rate to said server.